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**10th International Meeting
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Low Frequency Noise and Vibration and its Control
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**LOW-FREQUENCY TEST CHAMBER WITH LOUDSPEAKER ARRAYS
FOR HUMAN EXPOSURE TO SIMULATED FREE-FIELD CONDITIONS**

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ABSTRACT

The description of a test chamber used for infrasonic and low-frequency experiments, and the theoretical expected characteristics of the produced sound field are presented in this paper. The test chamber has been reconstructed based on an existing infrasound chamber. One of the objectives of this reconstruction has been to increase the frequency range where a flat frequency response can be obtained in a continuous three-dimensional listening region, which occupies a significant part of the volume of the chamber. In addition, the length of the impulse response in this region should be minimised as well. For this purpose, the use of active control has been considered. In the new test chamber, with inner dimensions of 2.72 m x 2.70 m x 2.40 m, a travelling plane wave will be generated; therefore, free-field conditions will be established. The acoustic plane wave will be produced by using a matrix of 4 x 5 loudspeakers in one of the walls. After travelling through the chamber, the sound will be absorbed on the opposite wall, which has a similar matrix of loudspeakers, by using active control. Theoretical results show that it is expected to reproduce recorded signals in free-field conditions within a frequency range of about 2 Hz to approximately 250 Hz. Deviations within ± 1.5 dB from a flat frequency response at any point in the listening zone are predicted in this frequency range. In addition, the impulse response will be a good approximation to a delayed delta function in that zone.

1. INTRODUCTION

For the study of noise, experiments are commonly carried out in a laboratory under well controlled conditions. Thus, to avoid the influence of undesired external factors in these experiments, a test enclosure is normally used where recorded signals are played back. It is desired that the reproduced sound should have practically the same characteristics as the one sensed by the microphone when the recording was made. Evidently, in the case of human exposure, the conditions of the exposure to hearing should be controlled.

However, when a recorded broadband signal is played back in a room, the reproduced sound undergoes a spectral colouration and reverberation due to the acoustic response of the room. The sound is also affected by the electroacoustic response of the reproduction system itself, mainly the loudspeakers.

If the wavelengths of the acoustic signals are relatively large compared to the dimensions of a test room, then the amplitude of the acoustic waves is practically independent of the position inside the enclosure. However, for experiments involving human exposure, a test chamber should be large enough to avoid the influence of being in a "test room".

A test chamber for infrasonic and low-frequency human exposure has been reconstructed based on an existing infrasonic chamber at the Acoustics Laboratory at Aalborg University, Denmark. It was decided to implement an active sound equalisation system in order to increase the frequency range where the test chamber can be used. Originally, the test chamber was design for experiments in the range of infrasound, where the maximum frequency intended to be reproduced was about 25 Hz. In this way, the dimensions of the chamber ensure that at this maximum frequency, the sound pressure is practically the same at any position within the chamber.

The active sound equalisation system is predicted to be able to compensate also for modifications in the original recorded signal produced by the loudspeakers and the reproduction system in general. The equalisation is based on simulation of a travelling plane wave inside the room. The listening zone is the part of the volume inside the test chamber separated from each of the two walls with the loudspeakers by 70 cm, limited by the two other walls and from the floor to the ceiling.

A preliminary implementation of the sound equalisation system by using 32 loudspeakers has shown that the simulation of a travelling plane wave inside the test chamber is feasible [1]. For that implementation eight input channels were used under the assumption of symmetrical conditions in the test chamber with respect to the middle, horizontal and vertical planes. The main restriction for the performance of the preliminary system was imposed by the arrangement of the loudspeakers. As a result, some loudspeakers were required to move very close to antiphase compared to others in the first wall, causing a high cancellation of sound around that wall and an inefficient transmission of sound across the chamber. Moreover, there was a significant difference in the amplitude of the signals fed to different channels. This caused that the displacements of the membranes of several loudspeakers were close to the limit of the linear behaviour while for other loudspeakers the displacements were very small. As a consequence of these two main effects, maximum sound pressure level obtained within the allowed limits of harmonic distortion was not sufficiently high.

For the rearrangement of the test chamber, the loudspeakers, 40 in total, were placed very close to the optimal position for the performance of the equalisation system. Compared to the results with the preliminary system, it is expected that higher sound pressure level will be achieved, the frequency range where sound equalisation is possible will be wider, and the frequency response at any point in the listening zone will be flatter.

In this paper, a description of the test chamber is presented. The implementation of the active sound equalisation system is analysed, and its expected final performance is also discussed. The resulting properties of the sound field have been predicted based on computer simulations.

2. TEST CHAMBER

2.1. Description

The test chamber is a rectangular room with dimensions 2.72 m wide (L_x), 2.70 m long (L_y) and 2.40 m high (L_z); a diagram of the chamber is shown in Fig 1. A set of 20 loudspeakers is placed in each of the two walls perpendicular to the y direction. Each of these walls consists of five wooden panels supported by a metal frame. The rest of the chamber is built of concrete. Behind each wooden wall there is a small room used as an enclosure for the loudspeakers. All the surfaces of these enclosures are covered with 100 mm mineral wool.

The 40 loudspeakers will be used to generate the acoustic waves inside the chamber. They are identical units (Seas 33 F-WKA) with a diameter of 13 inches.

To obtain a high sound insulation of the test chamber, it has been designed with a double concrete wall construction. Moreover, it was necessary to make the test chamber as airtight as possible in order that the chamber could be used at very low frequencies.

To reduce the reverberation time and attenuate external noise in the test chamber, the two concrete walls and the ceiling are covered with acoustic absorbent material, 50 mm thick, and the floor is covered with a carpet. In front of each wall with the loudspeakers there is a metal grid covered with fabric, which prevent the test subject from seeing the sound sources. A picture of the interior of the chamber can be seen in Fig 2. Here the screen that covers one wall with loudspeakers has been removed to show the arrangement of the transducers.

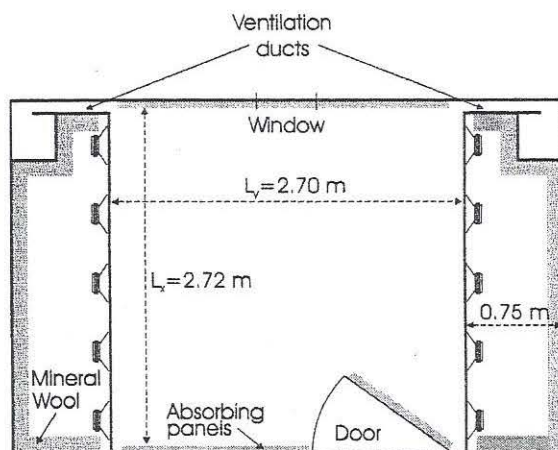


Fig. 1: Diagram of the test chamber seen from above.

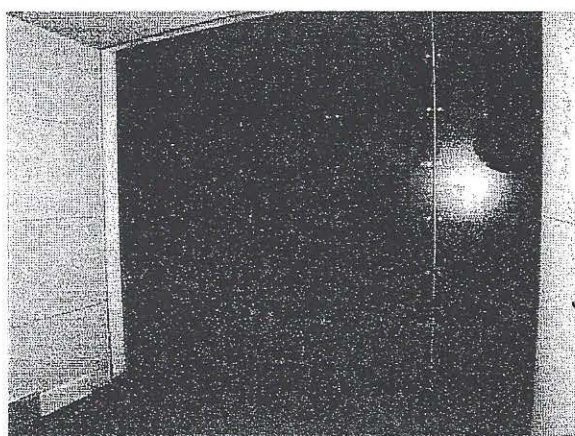


Fig. 2: View of one of the two walls with loudspeakers in the test chamber.

The test chamber is equipped with a ventilation system. In this way, experiments of long duration can be carried out. In addition, the temperature can be kept practically constant at about 21°C. There are two ducts, which are shown in Fig. 1, used by the ventilation system. The ducts are filled with filter material to eliminate the noise from the system and to regulate the flow of air. The openings of the ducts into the chamber are 0.08 m wide and 2.34 m high each.

2.2. Desired exposure conditions

From the acoustical point of view, there are several desired exposure conditions in the chamber: a) the test chamber should be isolated from external noise, b) the background noise inside the chamber should be minimised, c) it should be possible to obtain relatively high sound pressure levels, d) the harmonic distortions should be relatively low, e) a listening region large enough to allow for the movements of the listener is required, f) a flat frequency response should be obtained in the listening region, g) the duration of the impulse response in that region should be as short as possible.

As mentioned above, in order to isolate the chamber from external noise, this facility was built with hard materials, enclosed in a larger room, and the surfaces were covered with mineral wool. It was also desired to minimise the reverberation time in the chamber, in this way, any unwanted noise produced inside, for instance, by the movements of the listener, should be reduced. Therefore, the background noise in the chamber is expected to be relatively low.

At low frequencies, the human ear is less sensitive; therefore, it is desired to achieve relatively high sound pressure levels in the test chamber. At the same time, the harmonic distortion should be kept low. When the dimensions of the chamber are much smaller than the wavelength of the reproduced sound, the total pressure can be described by using the state equation for an adiabatic process: $pV^\gamma = \text{constant}$. In this way, the sound pressure Δp is given by

$$\Delta p = -\frac{\gamma p_o \Delta V}{V}, \quad (1)$$

where γ is the ratio of specific heats, p_o is the static pressure, and V is the volume of the chamber. As it can be seen, the sound pressure inside the chamber is proportional to the change in volume (ΔV) produced by the displacement of the membrane of the loudspeakers. Therefore, a loudspeaker with a large area and relatively large stroke should be used. In addition, the displacement of the membrane of the loudspeakers should be within the linear behaviour of these transducers. Because of these characteristics, the Seas 33 F-WKA loudspeakers were originally selected, and they were used after the rearrangement of the test chamber.

It can be observed from the reference threshold of hearing in the ISO 226 [2] that the maximum steepness of the threshold curve is approximately 30 dB per octave. Thus, in order to keep the effect from harmonic distortion products in the sound reproduction system inaudible for humans, the second order harmonic has to be attenuated 30 dB relative to the fundamental. The third order harmonic has to be attenuated 40 dB with respect to the fundamental and higher harmonics have to be

50 dB lower than the fundamental. This criterion was used to determine the highest sound pressure level expected to be reproduced in the chamber. Fig. 3 shows the SPL for the fundamental and the second and third harmonics produced by one of the SEAS loudspeaker inside a small enclosure of 184 litres. The loudspeaker was fed with a voltage of $2.5 V_{rms}$ at a single frequency. The excitation of the harmonics higher than the third one was practically negligible. According to the figure, it can be observed that this voltage of 2.5 V produced a harmonic distortion inside the enclosure within the limits of the criterion mentioned above. For this case, with a SPL of 129 dB the rms value of the displacement of the membrane of the loudspeaker was calculated to be 1.70 mm. This value was considered to be almost the limit for the maximum displacement of the loudspeakers in order to keep the harmonic distortion at a level sufficiently low. By considering this displacement of $1.70 mm_{rms}$, it is estimated that the maximum sound pressure level achieved by the loudspeakers in the test chamber in a pressure field condition, when all the loudspeakers are fed with the same signal, will be approximately equal to 123 dB (re. 20 μPa).

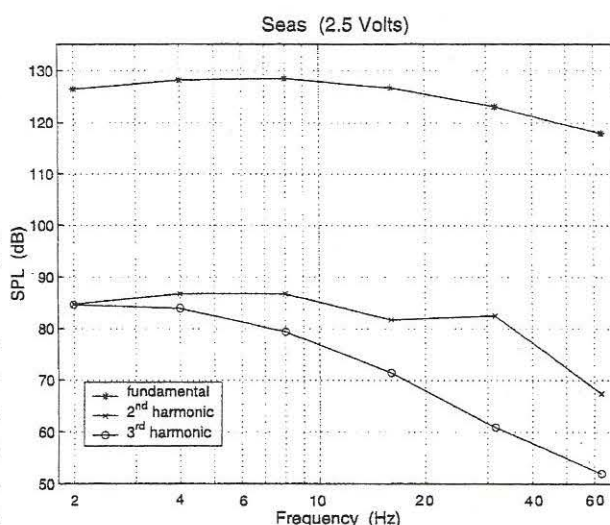


Fig. 3: SPL of the first three harmonics produced by one of the Seas loudspeakers inside an enclosure of 184 litres. The loudspeaker was fed with a pure tone of $2.5 V_{rms}$.

The sound pressure levels at very low frequencies will depend on the airtightness of the test chamber. Therefore, an attempt has been done to eliminate any leakage. It has been estimated that the lowest limiting frequency will depend mainly on the resistance to the flow of air in the ventilation system. With the ventilation system closed, it has been determined experimentally that the lower limiting frequency is 0.07 Hz.

In order to allow for the movements of the test subject during the experiments, the amplitude of the sound field should be independent of the position inside the listening region of the chamber. In certain cases, the subject e.g. might be asked to carry out a simple physical task. Under uncontrolled conditions, when the 40 loudspeakers are fed with the same input signal, the amplitude of the sound field inside the chamber will be practically the same at any position provided that the wavelength of the reproduced sound is much larger than the dimensions of the chamber. This is the case at frequencies lower than about 25 Hz. When the wavelength is similar or smaller than the dimensions of the chamber, the SPL is dependent on the position.

It is intended to reproduce not only pure tones, but also sound with various frequency components, for instance noise bands and environmental recorded signals. Therefore, the frequency response of the chamber should be flat over the frequency range of interest. At frequencies lower than about 25 Hz, the frequency response of the test chamber is expected to be considered practically flat. However, at higher

frequencies, the effect of the excitation of the different modes of the room becomes evident, and peaks and troughs appear in the curves of the frequency response.

In some applications, for instance in the reproduction of transient signals, the reverberation in the chamber is desired to be avoided. Therefore, the duration of the impulse response should be minimised. In the best situation, the impulse response of the room should be a good approximation to a delayed delta function. In this way, the reproduced sound will be a delayed version of the original sound from which the recording was made.

According to what is mentioned in the paragraphs above, the dimensions of the test chamber impose a limit in the highest frequency that can be reproduced. At frequencies higher than around 25 Hz the sound field will depend on the position, and the frequency response will contain peaks and troughs. Therefore, a possible implementation of a control system for the equalisation of the sound field was investigated. At low frequencies, passive means are not very efficient; therefore, active control appear to be an appropriate solution.

3. IMPROVEMENT OF EXPOSURE CONDITIONS

A sound equalisation system will be used in the test chamber to improve the exposure conditions. A theoretical investigation of this method of sound equalisation has been presented by Santillán [3]. The underlying idea is to generate an acoustic plane wave with the 20 loudspeakers in one of the walls of the chamber, and after this plane wave has travel across the chamber, the sound is absorbed at the opposite wall by using the other 20 loudspeakers in that wall. In order to generate a travelling plane wave, only the axial modes of the room to the direction of propagation of the plane wave should be excited. A factor that has a significant effect on the efficiency of this method of sound equalisation is the arrangement of the sound sources. They should be placed in the regions of minimum amplitude of those modes whose excitation should be avoided. The largest distance between the centre of two adjacent sources on the walls determines the maximum frequency that can be equalised.

For the implementation of the system of sound equalisation, 40 different channels will be used, one for each loudspeaker. Thus, the original signal to be reproduced will be convolved with 40 different FIR filters, one for each input channel. The resulting 40 filtered signals will be amplified and fed to the loudspeakers.

The aim for the implementation of the equalisation system is to determine the 40 different FIR control filters. The control filters are calculated analytically by a computer program. For this purpose, the sound field in the test chamber has to be estimated at a certain number of points (error sensor positions). To simulate a travelling plane wave, the error sensor positions should be located in two planes perpendicular to the direction of propagation of the simulated progressive wave. The distance between these two planes should be equal to the distance that the sound travels during one sampling period. In addition, the two planes should be placed in the centre of the chamber and the error sensor positions should be equidistant to each other. It has been estimated that an array of 6 x 6 error sensor positions is needed in each of the two planes to have an appropriate estimation of the sound field. Thus, the required information for the

calculation of the 40 FIR control filters consists of the different impulse responses from the input to each of the 40 channels and the output from each of the 72 error sensor positions.

By using the traditional least-squares approximation, the control filters are obtained by minimizing the error between a desired signal and the one received at each error sensor position. In order to minimise the length of the impulse response in the listening region, the desired signals are required to be delayed versions of the original input signal. The delays applied to the original signal to obtain the corresponding desired signals are proportional to the time of propagation of the simulated plane wave.

With the simulation of a travelling plane wave in the test chamber, it is expected to have a uniform spatial distribution of the sound pressure level. In addition, a relatively flat frequency response, and a very good approximation of the impulse response to a delayed delta function are predicted to be obtained. However, there is a restriction in the performance of the equalisation system at very low frequencies. This restriction can be made apparent by using an analogy between the simulation of a travelling wave in the test chamber and the propagation of a plane wave in an infinite duct with only one loudspeaker. For the latter, the sound pressure amplitude $p(x)$ at any position x in the duct produced by a monopole source at x_0 is given by

$$p(x) = q(x_0) \frac{\rho_0 c e^{-ik|x-x_0|}}{2S}, \quad (2)$$

where ρ_0 is the mean density of the medium, c is the sound speed, k is the wavenumber, S is the cross section of the duct, and q is the volume velocity of the monopole source. As it can be seen, the produced sound pressure amplitude will be proportional to the volume velocity of the sound source, which is the product of the velocity of the moving membrane of the source and its area. This means that in order to have the same sound pressure amplitude at different frequencies, the volume velocity of the sound source should be the same at different frequencies. This relation implies that the displacement of the surface of the monopole should be inversely proportional to the frequency. In this way, the displacement of the membrane of the loudspeakers should increase to infinity as the driving frequency approaches to zero in order to produce a travelling plane wave with its amplitude independent of the frequency. As mentioned above, the displacement of the membrane of the loudspeakers should be kept within the limit of the linear performance of the transducers to minimise the harmonic distortion. Therefore, a trade off exists between the lowest frequency of the travelling plane wave that can be established in the test chamber and the maximum sound pressure level that can be achieved.

An advantage of the simulation of a travelling plane wave in the test chamber is that the sound can be equalised in a significant part of the volume of the chamber. Only the region within 0.7 m from each of the two walls with the loudspeakers cannot be equalised properly. Thus, the expected region of equalisation will be the zone that occupies practically all the width of the room, from $y = 0.6$ m to $y = 2.1$ m, and from the floor to the ceiling. The resulting sound equalisation might, however, be affected in a region very close to the surfaces of the chamber (within approximately 10 cm) due to the presence of the absorbent material.

4. RESULTS

The characteristics of the expected sound field produced inside the test chamber by the simulation of a travelling plane wave have been predicted by using computer simulations. These predictions are expected to be good approximations to the final experimental results. The sound field inside the test chamber has been calculated by using a model in the frequency domain based on the excitation of the modes of the test chamber, and the loudspeakers have been considered as square pistons. For the calculations the sampling frequency has been assumed to be equal to 1 kHz.

As mentioned above, in the practical implementation of the sound equalisation system, the FIR control filters will be calculated analytically by a computer program. In fact, for that process, the considered sampling frequency will be 1 kHz. However, the implemented sound equalisation system will work by using a sampling frequency of 48 kHz; therefore, the input signal will be upsampled from 1 kHz to 48 kHz.

4.1. Sound pressure level

The expected sound pressure level in the test chamber has been estimated from computer simulation. Fig. 4 shows the rms value of the displacement of the membrane of two of the loudspeakers in the test chamber as a function of frequency to produce a SPL of 94 dB (re. 20 μ Pa) in the simulation of a travelling plane wave. One of the loudspeakers corresponds to the position in one of the corners of the wall and the other loudspeaker is considered to be placed in the centre of the wall. Here, it can be seen that the displacement of the membrane of the loudspeakers depends on the position of the sound source. In general, for the loudspeakers around the centre of the wall, the displacement is required to be higher.

The smallest displacement is demanded in the loudspeakers close to the corners. According to Fig. 4, to produce a SPL of 94 dB at 10 Hz, a displacement of the membrane of the loudspeaker in the centre of the wall is expected to be approximately equal to 0.414 mm_{rms}. With the criterion considered that the maximum displacement of the membrane of the loudspeakers should be 1.7 mm_{rms} to avoid undesired harmonic distortion, it is expected that the maximum SPL that can be obtained at 10 Hz will be approximately 106 dB. It can also be noticed in the graphs of Fig. 4 that the computer simulations predict that the displacement of the membranes of the loudspeakers needed to simulate a travelling plane wave with a given constant amplitude appears to be inversely proportional to frequency.

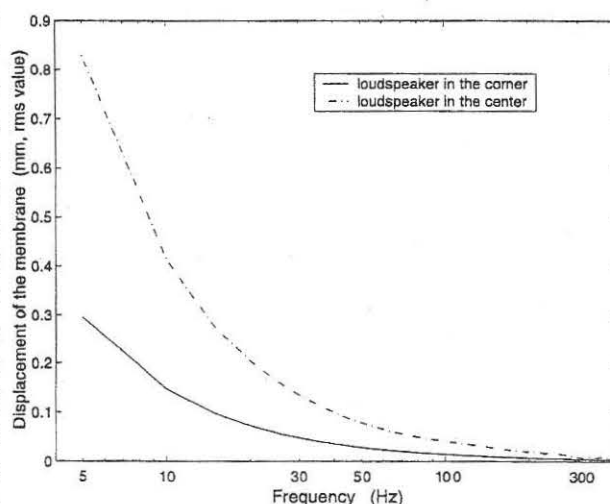


Fig. 4: Computer simulations of the displacement of the membrane of two loudspeakers in the simulation of a travelling plane wave in the test chamber. The plane wave is assumed to produce a SPL of 94 dB independent of the frequency.

4.2. Sound equalisation as a function of frequency

To evaluate the result of the expected sound equalisation in the listening region, an average error over M different positions between the obtained sound field and the desired one was calculated. This parameter, called in what follows the least-squares error, is given by

$$E_{LS} = \sqrt{\frac{1}{M} \sum_{m=1}^M (|p(\omega, r_m)| - |p_d(\omega, r_m)|)^2} \quad (3)$$

Here $p(\omega, r_m)$ is the sound pressure amplitude predicted to be reproduced at the position r_m , and $p_d(\omega, r_m)$ is the desired sound pressure amplitude at that position. The least-squares error calculated by using $7 \times 7 \times 6$ positions distributed equidistantly in the entire listening region is shown in Fig. 5. Here, the responses of the 40 control FIR filters, calculated from computer simulations in the time domain, were transformed into the frequency domain; then these frequency responses were used to determine the least-squares error as a function of the frequency. As a criterion for an appropriate good sound equalisation, it has been considered that deviations in the reproduced sound pressure level within ± 2 dB from the desired value should be obtained in practically the complete listening region. It has been observed, that this criterion is achieved if the least-squares error is less than 0.3. Therefore, it can be seen in Fig. 5 that with the implementation of the control system a satisfactory sound equalisation is expected up to approximately 270 Hz in the complete listening region. According to the graph in Fig. 5, the result is improved in the frequency range up to approximately 250 Hz. In this frequency range, the expected deviations in the produced sound level from the desired values are within ± 1.5 dB in almost the complete listening zone.

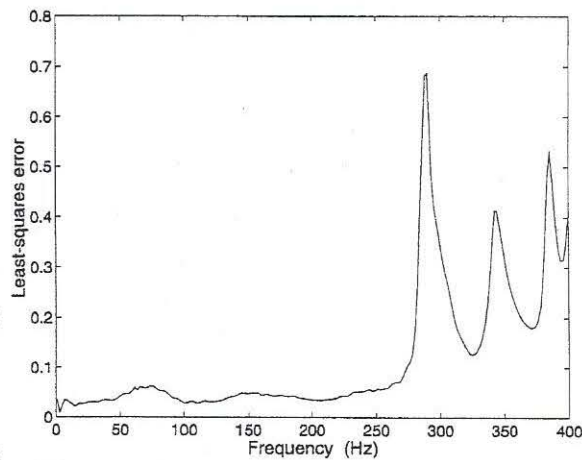


Fig. 5: Least-squares error as a function of frequency. For values less than 0.3 of this parameter, deviations of the SPL in the complete listening region are expected to be within ± 2 dB from the desired level.

4.3. Impulse and frequency responses

As an example of the expected performance of the sound equalisation system, the impulse responses at four different positions and their corresponding frequency responses are shown in Fig. 6. The curves before and after equalisation can be compared in this figure, where the responses before equalisation has been obtained assuming that the 40 loudspeakers are fed with the same signal. It should be noticed that these positions are different from the error sensor positions. The impulse responses after the application of the control filters are depicted with an offset of -1.5 units for clarity, and the curves in the frequency domain have been normalised in order that the desired sound pressure level corresponds to zero dB. It can be seen in the

figure that the frequency responses have been significantly improved with the application of the equalisation system. The peaks and troughs that are present in the curves before equalisation have been considerably removed after the simulation of a travelling plane wave. The frequency responses after the application of the equalisation system can be considered flat up to approximately 260 Hz, which agrees with the result from the last section. It can also be noticed that at frequencies higher than 260 Hz, the frequency responses after equalisation have been improved as well compared to the original curves, but there are relatively significant deviations from the desired value. From the curves of the impulse responses, it can be noticed that the lengths of the original impulse responses have been decreased with the application of the control filters, and the resulting impulse responses are good approximation to a delayed delta function. The differences between the impulse responses after equalisation and a delayed delta function are due mainly to the performance of the system at frequencies higher than 260 Hz, where it is not possible to obtain a good sound equalisation.

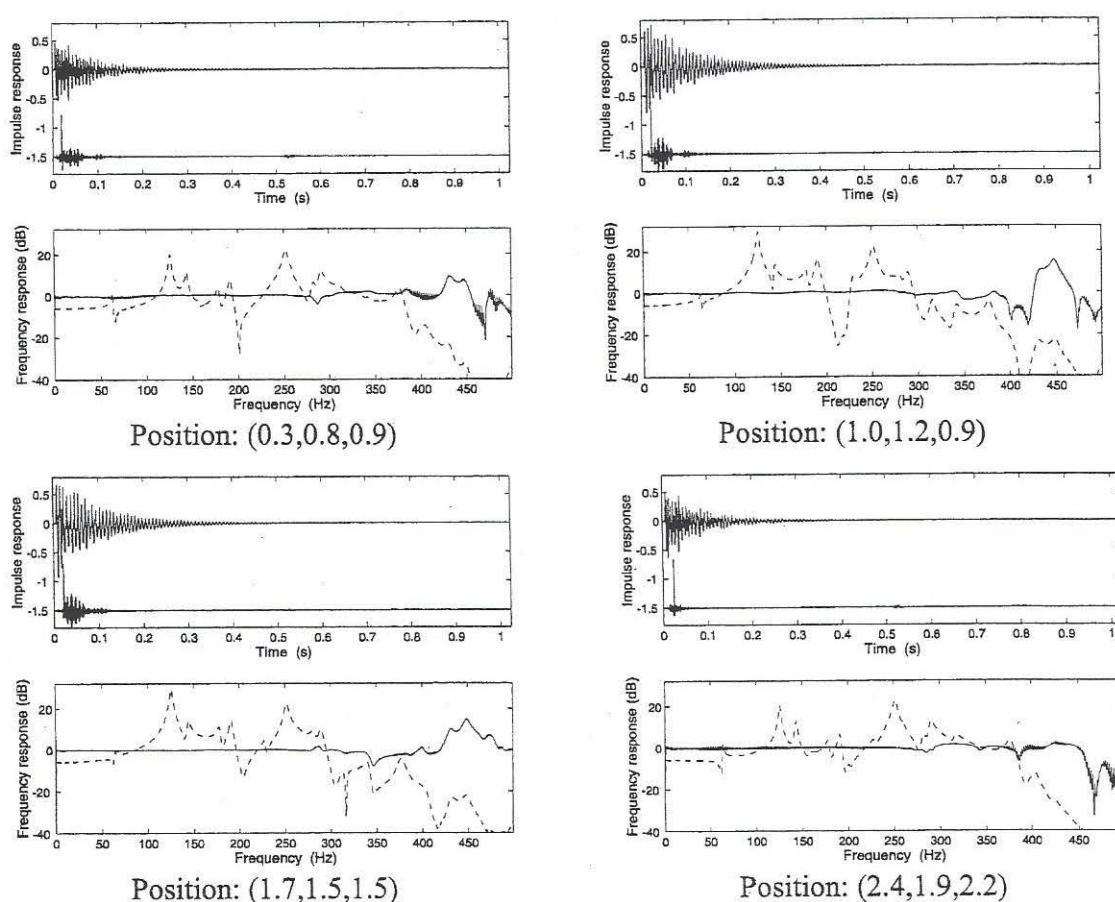


Fig. 6: Impulse and frequency responses at four different positions before and after sound equalisation. The impulse responses after equalisation are shown with an offset of minus 1.5 unit for clarity. The solid lines in the frequency responses represent the results after equalisation and the dashed lines correspond to the results before equalisation, the last ones include the response of an anti-aliasing filter used in the simulations.

4.4. Spatial uniformity of the SPL

As an example of the spatial distribution of the sound pressure at a single frequency, Fig. 7 and Fig 8 show the sound pressure level in the plane $z = 1.35$ m for a driving frequency of 250 Hz before and after sound equalisation respectively. Before equalisation, the sound field is produced by the 40 loudspeakers moving in phase. Here it can be observed the axial modes to the y direction are mainly excited since a pattern of standing wave in that direction appears to be formed. This is a consequence of distribution of the loudspeakers. However, the sound pressure amplitude depends on the position along the y axis and high differences can be seen. In contrast, the sound pressure level is practically constant in the complete listening zone after equalisation as it can be noticed in Fig. 8. The deviations of the obtained sound pressure level from the desired value of zero dB are relatively high within a distance of 0.7 m from the loudspeakers though. It should be mentioned that after equalisation the sound pressure level in the listening zone is predicted to be almost independent of the position in the z direction; the spatial distribution of the sound field in any plane xy is basically the same as the one shown in figure 8.

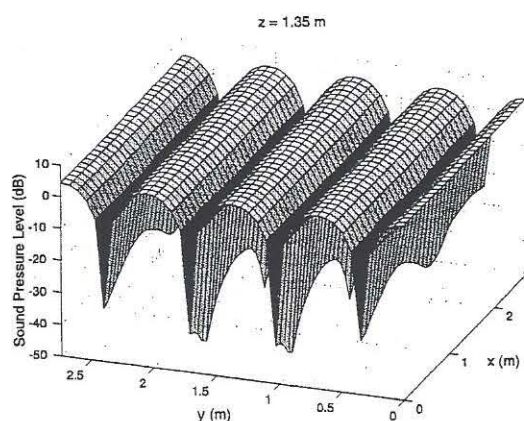


Fig. 7: Distribution of the sound pressure level at the frequency of 250 Hz before sound equalisation. The graph corresponds to the sound field in the plane $z = 1.35$ m produced when the 40 loudspeakers are fed with the same signal.

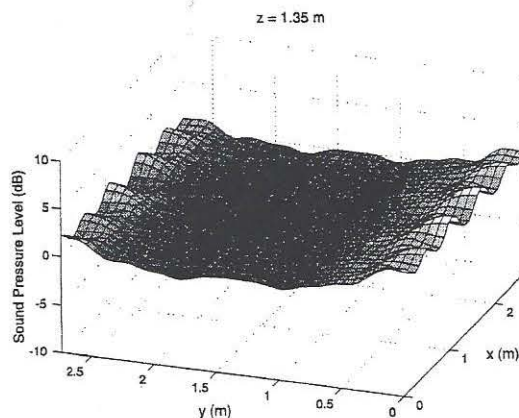


Fig. 8: Sound pressure level in the plane $z = 1.35$ m after equalisation. The driving frequency is 250 Hz.

5 DISCUSSION

It has been shown that a good sound equalisation can be obtained in the test chamber by the simulation of a travelling plane wave. Here the sound propagates with a negligible effect from the walls, this means, free-field conditions are simulated. This result might offer another application of the proposed sound equalisation system since free-field conditions at very low frequencies are difficult to obtain in a common anechoic chamber. The inner dimensions of the anechoic chamber have to be much larger than the wavelength of the sound to be reproduced. Moreover, absorbing material covering

the interior of the room needs to be relatively large, at least one quarter of a wavelength. Such dimensions become significantly large at very low frequencies.

6. CONCLUSIONS

According to the presented theoretical results based on computer simulations, it has been shown that an active sound equalisation system can be used to improve the exposure conditions in a test chamber at infrasonic and low frequencies. These results are also supported by a previous preliminary implementation of the control system. The sound equalisation is based on the simulation of a travelling plane wave inside the test chamber, where 40 loudspeakers are used to produce the sound field. It can be concluded that for the considered test chamber a flat frequency response is predicted to be obtained up to approximately 250 Hz, where deviations within ± 1.5 dB in the SPL of the reproduced sound field from the desired value are expected. The zone where sound equalisation is possible is a continuous three-dimensional region that occupies a significant part of the interior volume of the test chamber. In addition, the impulse response in this listening region will approximate a delayed delta function in the considered frequency range.

The performance of the sound equalisation system at the low end of the frequency range will be limited by the allowed displacement of the membrane of the loudspeakers. The required displacements of the membranes vary inversely proportional to the frequency. In order to keep the harmonic distortions within an inaudible level, it is expected that at 10 Hz the maximum SPL that can be achieved will be 106 dB (re. 20 μ Pa).

In the region of infrasound, a pressure field can be produced, where the maximum predicted SPL is 123 dB (re. 20 μ Pa). The lowest frequency that can be achieved in this case will be determined by the airtightness of the chamber, mainly due to the effect of the airconditioning system. With the airconditioning system closed, the lowest limiting frequency is 0.07 Hz according to the measurements that have been carried out. In experiments where it is necessary to reproduce frequencies in the frequency range between 0.07 Hz up to 270 Hz, the proposed equalisation system by the simulation of a travelling wave and the pressure field conditions can be combined to include the total frequency range.

7. ACKNOWLEDGMENTS

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